## TITLE OF THE INVENTION

# MESSAGE CONVERSION SERVER AND IP TELEPHONE

## BACKGROUND OF THE INVENTION

### Field of the Invention

The present invention relates to a calling method used when a caller makes a call to a callee in a one-to-one communication mode on the Internet.

Non-patent documents listed below are referred in the description in a following paragraph.

[Non-patent document 1]

IETF RFC3261

[Non-patent document 2]

IETF RFC2327

[Non-patent document 3]

IETF RFC1889

In recent years, a tendency to integrate exchange-based telephone communications networks into IP networks has been developed with a rapid advance in IP network technology. Telephone communications carriers have a plan to transfer data under voice (hereafter, simply referred to as DUV) on their own IP networks through Voice over IP (hereafter, simply referred to as VoIP). VoIP consists of two protocols, one controlling call signaling and sessions and the other controlling DUV transfer. The

Internet Engineering Task Force (hereafter, simply referred to as IETF) made a specifications of Session Initiation Protocol (hereafter, simply referred to as SIP), IETF RFC3261, defining the method for call signaling and sessions. For example, Session Description Protocol (SDP), IETF RFC2327, is applicable to the description of a session including an agreement on a codec and transfer rate to be internally used in SIP.

Although no specific DUV protocol is defined in SIP, Real-time Transport Protocol (RTP), RFC1889, is commonly used.

According to IETF RFC3261, SIP is such a protocol that a SIP message consisting of a SIP Start Line, a SIP message header, and a SIP message body is sent/received between two calling parties via a SIP server to make an agreement about the call signaling mode on the callee side, the voice, image protocol and bit rate to be used during conversation. Generally, the SIP Start Line describes the behavior of the message originator, the SIP message header describes the telephone number of a callee, the SIP server passed, and Call-ID (call origination administration number), and the SIP message body describes the proposed voice, image protocol, and bit rate to be used during conversation.

Now, a procedure is described in brief ranging from the start to end of a two-party conversation through SIP

described in IETF RFC3261 and the problems with the procedure are explained.

Fig. 1 is a network diagram showing two-party conversation through SIP.

Fig. 2 is a sequence diagram explaining a flow of two-party conversation through SIP.

In Fig. 1, UserA, who belongs to a domain 3-1 and has an IP Telephone 2-1, makes a call to UserB, who belongs to a domain 3-3 and has an IP Telephone 2-2 via SIP servers 1-1 to 1-3.

First, UserA sends a Start Line INVlTE and a SIP message for UserB to the SIP server A1-1 to establish a call with UserB (11). The SIP server A1-1, when receiving the INVITE message, adds a VIA header to its message header and transfers the SIP message to the SIP server B1-2. At that time, it also sends the SIP message containing a Start Line 100Trying to the IP Telephone 2-1, which is the callee (destination) of the message (12). The SIP servers B1-2 and C1-3, when receiving the SIP message, take the same actions and transfer the message to the UserB's IP Telephone 2-2.

The IP Telephone 2-2, when receiving the SIP message, sends a Start Line 180Ringing and a SIP message for UserA to the SIP server C1-3 (13) to sound a ringing tone on the UserB side. The SIP message containing a Start Line

180Ringing terminates at the IP Telephone 2-1 via the SIP server.

The IP Telephone 2-2, when UserB picks up a telephone receiver, sends a Start Line 200OK and a SIP message for UserA (14), which in turn, terminates at the IP Telephone 2-1 via the SIP server. The IP Telephone 2-1 send back an acknowledge (ACK) signal in response to the message (15) and when the ACK is received, a voice packet passes through a main signal path, enabling the two parties to initiate a conversation between them (16).

At the end of the conversation, UserA's IP Telephone 2-1 sends a Start Line BYE and a SIP message for UserB (17), which in turn, determines at the IP Telephone 2-2 via the SIP server. In response to the message, the IP Telephone 2-2 sends back the ACK to the IP Telephone 2-1 via the SIP server (18). When the ACK is received, the conversation ends.

SIP is a protocol for sending and receiving SIP messages between a caller and a callee. A UserID and its DomainID are described in the headers, From specifying the caller contained in the message header, To specifying the callee, and Via specifying the SIP servers passed (proxy mode) because these information can be delivered as they are when the callee sends it back. To establish a session with

the callee through SIP, the caller describes the IP address of his/her own terminal or assigned DomainID in the headers.

With regard to VoIP, a protocol which informs the callee of no UserID (e.g. telephone number or UserID) of the caller, such a mode has been proposed in IETF RFC3261 that the caller terminal creates a random UserID, registers the random UserID and the IP address of the terminal in the SIP server, and originates a call with the random UserID designated as the caller. Through this protocol, the whole procedure for making a Caller Anonymous Call is performed on the caller side.

If the IETF-supported communication mode, in which the caller makes a Caller Anonymous Call with his/her UserID concealed, is used in the SIP system, only the random UserID is registered in the SIP server. For this reason, the caller would make a Caller Anonymous Call not only to the callee but also the SIP server at the same time. The SIP server is difficult to control and manage calls because it cannot guess the real UserID from the UserID registered in it. In the IP telephone services provided by communications carriers and others, user management is required, for example, caller identification, service permission/denial determination, and talk time management.

If a malicious third party intercepts a SIP message, he/she can know the caller, the caller's SIP-URL, and the

assigned DomainID described in the SIP message, causing damages such as nuisance calls.

In addition, if the malicious third party knows the IP address of the IP Telephone, he/she can transfer a vast amount of packets to the IP Telephone after the end of conversation, making an attack, for example, DOS (Denial of Service) disturbing processing on equipment at a high possibility.

SUMMARY OF THE INVENTION

An object of the present invention is to provide a method "Non caller informed call", which enables the SIP servers to manage caller information (UserID identifying the UserID of the caller sending the SIP message and his/her DomainID) while concealing the information from the callee and the malicious third party.

Another object of the present invention is to make it difficult for the third party to identify the IP address assigned to the IP Telephone to minimize any damage caused when he know the IP address.

According to one aspect of the present invention, a packet forwarding device, transmits a message sent by a caller to a specified callee, wherein the device has a processing part for providing at least either the function for converting or the function for erasing at least part of

the message sent by the caller upon his/her request and a control part for determining whether at least the part of the message should be converted or erased and converts or erases at least the part of the message based on the determination in the control part. This mechanism enables information identifying the caller to be concealed from the callee.

At least the part of the message converted or erased as described above may be one of or any combination of:

(1) the part identifying the user on a calling side in the SIP message header on an IP packet payload containing the message,

(2) the part identifying the caller's domain in the SIP message header on the IP packet payload containing the message,

(3) the part of a Via tag in the SIP message header on the IP packet payload containing the message,

(4) the part indicating the Call-ID domain in the SIP message header on the IP packet payload containing the message, and

(5) the part identifying a UserID in the SIP message body on the IP packed payload containing the message.

It may be possible that the control part analyzes the content of the message, when receiving it, and if any given character string or header is detected, the processing part converts or erase part of the message. Any given character

string is a series of numeric characters contained in the position of the first numeric string, for example "184". Any given header is the SIP message header and when its extended header is detected, the above processing part may convert or erase part of the message.

Additionally, it is preferable that a table, in which the correspondence between the contents of part of the message before and after conversion is contained, is prepared. According to one aspect of the present invention, to conceal information on SIP-URL and the assigned domain of the caller, the SIP server installed at a relay point between the caller and the callee converts the SIP message. The SIP server with the message conversion function is characterized that it provides the method for converting or erasing part of the message originated by the caller, the processing part for determining whether it should be converted or erased or not, the processing part for determining information to be used in conversion, and the table containing the rule of conversion.

According to the aspect of the present invention, to make it difficult for the third party to know the IP address of a caller's IP Telephone, the IP Telephone creates or obtains a temporary IP address to be used only once when the SIP message is sent and discards it as soon as the conversation ends up. The IP Telephone is characterized in

that it provides the method for creating or obtaining the IP address in conjunction with the origination of the SIP message and discarding it as soon as the conversation ends, the processing part for registering the temporary IP address in the SIP server, the processing part for canceling its registration from the SIP server, and the processing part for creating a random interface ID.

It is possible that optionally, the step for converting the SIP message using the SIP server and/or the step for obtaining the temporary IP address using the IP Telephone may be used. According to another aspect of the present invention, the SIP telephone communication method involves a step for receiving the SIP message, a step for checking the SIP message for any request for Anonymous Call, a step for executing at least one of two operations, modification and deletion on at least part of the SIP message if any request is detected, and a step for sending the SIP message processed as described above. It is preferable that if the request for Anonymous Call is detected, modification is made on at least part of the SIP and a table containing the correspondence between the contents before and after modification is created.

A second another aspect of the present invention, the IP telephone communication method involves a process for modifying the caller address to its temporary address at the

initiation of conversation and the process for discarding the above temporary address at the end of conversation.

A third another aspect of the present invention, the method for converting or erasing part of the message sent by the caller upon his/her request is characterized by a step for determining whether part of the message should be converted or erased, a step for determining information to be used in conversion if it is determined to be modified, and a rule applicable to conversion. In addition, a telephone set is characterized in it involves a step for modifying the address of a caller in conjunction with origination of the message every time he/she makes a call to prevent it from being disclosed and a step for disposing the address indicating the different UserID used at conversation as soon as the conversation ends, and further provides the method for assigning the addresses indicating different UserIDs at call origination and call receiving and the method for sending the user's call. The scope of the present invention includes the methods, devices, and systems described above.

As explained above, the invention achieves the function compatible with the exchange-based Anonymous Call function in the IP Telephone.

BRIEF DESCRIPTION OF THE DRAWINGS

Fig. 1 is a schematic view showing an example of the SIP network created for explaining the object of the present invention;

Fig.2 is a sequence diagram showing an example of the procedure for making a call in the SIP network created for explaining the object of the present invention;

Fig. 3 is a schematic drawing showing the SIP network of the present invention;

Fig. 4 is a flow chart showing the operational principle of the SIP server with the message exchange function of the present invention;

Fig. 5 is a block diagram of the SIP server with the message exchange function of the present invention;

Fig. 6 is a block diagram of the SIP server with the message exchange function of the present invention;

Fig. 7 is a view showing the format of the IP packet containing the SIP message;

Fig. 8 is a schematic view showing an example of the network using the SIP server with the message conversion function of the present invention.

Fig. 9 is a sequence showing the procedure for making a call in the example of the network using the SIP server with message conversion function of the present invention;

Fig. 10 is a view showing the contents of a SIP message header unconverted and converted by the SIP server with the message conversion function of the present invention;

Fig. 11 is a view of the content of a SIP message body (SDP) unconverted by the SIP server with the message conversion function of the present invention;

Fig. 12 is a view of the content of a SIP message body (SDP) converted by the SIP server with the message conversion function of the present invention;

Fig. 13 is a view showing the conversion table stored on the SIP server with the message conversion function of the present invention;

Fig. 14 is another view showing the conversion table stored on the SIP server with the message conversion function of the present invention;

Fig. 15 is a schematic view of the network using the SIP server with the message conversion function of the present invention;

Fig. 16 is a sequence diagram showing the procedure for making a call in the network using the SIP server with the message conversion function of the present invention;

Fig. 17 is a view showing the contents of the SIP message header unconverted and converted by the SIP server with the message conversion function of the present invention;

Fig. 18 is a view showing the conversion table stored on the SIP server with the message conversion function of the present invention;

Fig. 19 is another view showing the conversion table stored on the SIP server with the message conversion table of the present invention;

Fig. 20 is a schematic view showing the network using the SIP server with the message conversion function of the present invention;

Fig. 21 is a sequence diagram showing the procedure for making a call in the network using the SIP server with the message conversion function of the present invention;

Fig. 22 is a view showing the contents of the SIP message unconverted and converted by the SIP server with the message conversion function of the present invention;

Fig. 23 is a view showing the conversion table stored on the SIP server with the message conversion function of the present invention;

Fig. 24 is another view showing the conversion table stored on the SIP server with the message conversion function of the present invention;

Fig. 25 is a view showing the network using the SIP server with the message conversion function of the present invention;

Fig. 26 is a sequence drawing showing the procedure for making a call in the network using the SIP server with the message conversion function of the present invention;

Fig. 27 is a view showing the contents of the SIP message header unconverted and converted by the SIP server with the message conversion function of the present invention;

Fig. 28 is a flow chart showing the principle of the operation of the IP Telephone of the present invention ranging from the step for obtaining the temporary address to the step for discarding it;

Fig. 29 is a block diagram showing the function of the IP Telephone of the present invention;

Fig. 30 is a systematic diagram of IPv6 addresses; and

Fig. 31 is a view showing the content of the SIP message header when a Caller Anonymous Call is made.

DETAILED OF THE PREFERRED EMBODIMENT

Embodiment 1

Fig. 3 is a view showing a first embodiment of an IP telephone line network using a SIP server with a message exchange function of the present invention.

Fig. 4 is a flow chart explaining an operational procedure of the SIP server 12-1 with the message exchange function.

Fig. 5 is a block diagram explaining the function of the SIP server with the message exchange function 12-1.

Fig. 6 is a view showing the hardware configuration of the SIP server with the message exchange function 12-1.

Fig. 7 is a view showing an IP packet containing a SIP message. The IP packet is represented by 60, an IPv4/v6 header by 61, a TCP/UDP header by 62, a SIP Start Line by 64, a SIP message header by 65, and a SIP message body (SDP) by 66.

Now, referring to Fig. 4 and Fig. 5, the operational principle of the SIP server 12-1 with the message exchange function is described below.

First of all, The IP packet indicated in Fig. 7 via IF 51. Second, receiving of the SIP massage 22 is performed at a message sending/receiving part 45 and Start Line check 23, Header check 24, and body check 25 are performed at a SIP message checking part 44. If any error is detected in the SIP message, the process ends with an error response notification 33 issued.

If no error is detected, then message conversion check 26 is performed at message conversion request check 26 at a message conversion/processing part 47.

If no conversion request is detected, a Via header in described in the SIP message header 64 at the exchange/processing part 47 and then the SIP message is sent

via IF51. If the conversion request is issued with a flag 561 and 571 (see Fig.31) indicating that the "No caller informed call" is to be originated, the step for converting the message header conversion 27 and the step for converting the message body 28 are performed at the message conversion/processing part 47. These steps conceal information on the caller from its callee (destination). After the conversion, a step for extraction converted information 29 is performed at a converted information extraction/transfer part 46 to pick up header body information necessary for creating a conversion table. Then, a step for creating the conversion information table 30 is performed at a converted entry creation/modification part 49, writing into the conversion information table 31 is performed at a converted entry I/O part 48, and the converted entry is registered at a converted entry registration part 50. The converted SIP message undergoes message transfer 32 at a SIP message sending/receiving part 45 via IF51.

According to the embodiment of the present invention, the function of the converted entry creation/processing part 49 in the SIP message processing part 41 and the conversion table processing part 42 shown in Fig. 5 is executed on CPU72 shown in Fig. 6. The function of the converted entry I/O part 48 shown in Fig. 5 is executed at

a conversion table fetching part 75 shown in Fig. 6. The function of the conversion table registration part 43 shown in Fig. 5 is executed at a conversion table storage part 74 shown in Fig. 6.

Embodiment 2

Now, a more detailed embodiment is described below.

Fig. 8 is a schematic view of a network configuration showing the second embodiment of the SIP server of the present invention.

Fig. 9 is a sequence diagram explaining the communication procedure in the embodiment of the network using the SIP server with the message conversion function of the present invention.

In the second embodiment, UserA makes a Caller Anonymous Call to UserB. The step for processing the Non caller informed call is performed on the SIP server A12-2 with the message exchange function. The step for converting the SIP message is performed on the SIP server A (12-1) with a header conversion function used by the caller in sending the SIP message at the steps 111 and 112 in the sequence ranging from the start to the end of conversation as shown in Fig. 9. Note that the SIP server with the conversion function behaves as shown in the first embodiment. Fig. 10,

Fig. 11, and Fig. 12 show the content of the converted SIP message in the second embodiment.

Fig. 10 is a view showing the header part of the SIP message (65 in Fig. 7). In the upper part, the unconverted header is shown and in the lower part, the converted header is shown. In the second embodiment, a UserID 142 of the From tag and a part identifying the user of SIP-URL 143 in the unconverted header 141 are converted into, for example, character strings 147 and 148 such as Anonymous, from which the UserID of the caller cannot be guessed.

Fig. 11 is a view showing a body part of the SIP message (Unconverted). The body part of the SIP message is represented by 66 in Fig. 7.

Fig. 12 is a view showing the body part of the SIP message (converted). In this figure, the part identifying a UserID 152 of the message body (SDP) 151 is converted into 156. the part 153 identifying the user's address to be used during conversation in an "o","c" tag is converted into the IP address if described in FQDN (Fully Qualified DomainID).

As described above, the prevent invention enables information on the caller to be concealed by converting information on the caller, based on which the callee identify the UserID of the caller including the UserID 142 of the From tag, the part identifying the SIP-URL user 143,

and the part 152 identifying the UserID of the message body (SDP).

Fig. 13 and Fig. 14 are views showing the conversion tables stored on the SIP server 12 with the conversion function in the first embodiment of the present invention. These conversion tables include the table 170, which associates all the UserIDs converted into the same Call-ID 172 with their real UserIDs 171, the table 180, which associates anonymous UserIDs 181 previously stored on the SIP server as "Anonymous1-100" with their real UserIDs 182, and others. Reference to these tables associating anonymous UserIDs 181 with Call-ID 172 makes successful routing of the SIP messages.

Embodiment 3

Fig. 15 is a schematic view showing the network configuration in the third embodiment of the SIP server with the message conversion function of the present invention. Unlike the embodiment shown in Fig. 8, anonymous processing is executed on the SIP server B.

Fig. 16 is a sequence diagram showing the procedure for processing conversation on the network using the SIP with the message exchange function of the present invention.

In the third embodiment of the present invention, UserA makes a Callee Anonymous Call to UserB. The step for

converting the SIP message is performed on the SIP server A (12-3) with a header conversion function A, which sends the SIP message at the steps 211 and 212 in the sequence ranging from the start to the end of conversation as shown in Fig. 16. Note that the SIP server with the conversion function behaves as shown in the first embodiment.

Fig. 17 is a view showing the content of the SIP message converted on the SIP server with the message conversion function. Both of the unconverted and converted message headers are shown (65 in Fig. 7).

In the third embodiment of the present invention, the Via tag 222 is erased and only the Via tag 232 of an own server is described in the unconverted message header 221. The UserID 223 in the From tag is converted into 233, from which no UserID of the callee can be guessed, such as Anonymous and the part 224 identifying SIP-URL is converted into 225, from which no UserID and its domain can be guessed. In addition, the part 225 identifying the Call-ID's domain is converted into 235. The rule of message body (SDP) conversion is the same as that of the first embodiment of the present invention.

As described above, in the embodiment shown in Fig. 17, the Via tag indicating the relay point for the message can be deleted to prevent the call source from being guessed. Alternately, the part 225 identifying the Call-ID's domain

can be converted into temporary DomainID 235, from which no domain can be guessed.

Fig. 18 and Fig. 19 are views showing the conversion tables stored on the SIP server 12-3 with the conversion function in the second embodiment of the present invention. These conversion tables include the table 271, which associates all the UserIDs converted into the same Call-ID 273 with their real UserIDs 272, the table 281, which associates anonymous UserIDs 282 previously stored on the SIP server as "Anonymous1-100" with their real UserIDs 283, and others. The use of these tables in routing SIP messages conceals information from the callee, ensuring information security.

Embodiment 4

Fig. 20 is a view showing the network configuration of the SIP server in the fourth embodiment of the present invention.

Fig. 21 is a sequence diagram showing the procedure for processing conversation in the network using the SIP server with the message conversion function.

In the fourth embodiment of the present invention, UserA makes a Caller Anonymous Call. The step for converting the SIP message is performed on the SIP server A (12-4) with the header conversion function used by the caller in sending

the SIP message at the steps 311 and 314 and on the SIP server C (12-5) with the header conversion function at the steps 312 and 313, which sends the SIP message to the callee in the sequence ranging from the start to the end of conversation as shown in Fig. 21. Note that the SIP server with the conversion function behaves as shown in the first embodiment.

Fig. 22 is a view showing the content of the SIP message converted on the SIP server with the message conversion function. In the fourth embodiment of the present invention, first, the UserIDs 322 and 323 of the From tag in the unconverted header 321 is converted into character strings 326 and 327, from which no UserID of the callee can be guessed such as Anonymous and the message body (SDP) is converted in accordance with the same rule as that of the first embodiment of the present invention. Second, the Via tag 332 in the unconverted header 331 is erased on the SIP server 12-5 and only the Via tag 336 of an own server is described. The part 333 identifying SIP-URL in the From tag is converted into 337, from which no DomainID can be guessed. The part 334 identifying the Call-ID's domain is converted into 338.

Fig. 23 is a view showing the conversion table stored on the SIP server 12-4 with the conversion function.

Fig. 24 is a view showing the conversion table stored on the SIP server 12-5 with the conversion function.

The conversion tables stored on the SIP server 12-4 include the table 341, which associates all the UserIDs converted into the same Call-ID 343 with their real UserIDs 342, the table 351, which associates anonymous UserIDs 352 previously stored on the SIP server as "Anonymous1-100" with their real UserIDs 353, and others. The conversion tables stored on the SIP server 12-5 include the table 361, which associates Call-ID 362 with DomainIDs 364, and others.

Embodiment 5

Fig. 25 is a schematic view showing the network structure in the fifth embodiment of the SIP server of the prevent invention.

Fig. 26 is a sequence diagram showing the procedure for processing conversation in the network using the SIP server with the message conversion function of the present invention.

In the fifth embodiment of the present invention, UserA make a Caller Anonymous Call to UserB. The step for converting the SIP message is performed on the SIP server A (12-6) with a header conversion function belonging to the top level domain for each communications carrier at the

steps 411 and 412 in the sequence ranging from the start to the end of conversation as shown in Fig. 26.

Fig. 27 is a view showing the content of the SIP message converted on the SIP server with the message conversion function in the fifth embodiment of the present invention. In the fifth embodiment of the present invention, the Via tag 421 is erased and only the Via tag 426 of an own server, the communication carrier, is described in the unconverted message header 420. The UserID 422 and the part 423 identifying user's SIP-URL in the From tag are converted into character strings 427 and 428, from which no UserID of the callee can be guessed, such as Anonymous. In addition, the part 424 identifying the caller's domain in the Call-ID tag is converted into the top level domain 429 of the communication carrier. The rule of message body (SDP) conversion is the same as that of the first embodiment of the present invention. The conversion tables stored on the SIP server 12-6 with the conversion function in the fourth embodiment of the present invention are shown in Fig. 18 and Fig. 19 and the contents of these tables are the same in those of the second embodiment of the present invention.

Embodiment 6

Fig. 28 is s flow chart explaining the operational principle of the IP Telephone using the temporary IP address at conversation.

Fig. 29 is a block diagram explaining the function of the IP Telephone. The operational principle of the IP Telephone in the sixth embodiment of the present invention is described below.

Fig. 30 is a systematic diagram of IPv6 addresses.

Fig. 31 is a view showing the SIP message header used when a Caller Anonymous Call is made.

First, the procedure for initiating a call is described below. When the caller originates a call to another user, the IP Telephone 521 initiates the step for sending the SIP message and executes the step for selecting address acquisition method 502. If a random address creation method is selected, then the step for sending Router Solicitation 503 is performed to obtain an IPv6 address prefix 551 from a router in the same subnet. When the router sends Router Advertisement in response to Router Solicitation, a step for receiving Router Advertisement 505 is performed to obtain the address prefix 551.

Second, a step for creating an interface ID 506 is performed at the random interface ID creation part 528 to create an IPv6 interface ID552.

The examples of the address prefix and the interface ID are represented by 553 and 554 in Fig. 30.

Third, a step for creating an IP address 507 is performed at a temporary IP address processing part 529 using the address prefix 501 and the interface ID552.

If the option of acquisition from the DHCP server is selected at the step for selecting the address acquisition method 502, a request for address acquisition 504 is issued to any address distribution server, for example, the DHCP server to execute a step for obtaining the temporary IP address 508.

Whenever an IP call is made through IPv4, the address should be obtained from an external server..

Fourth, the modified entry or new registration entry of user information is crested at a user data processing part 532 using the temporary address and the UserID to execute a step for registering the user's account 509.

Fifth, a step for creating the SIP message 510 at a SIP message header creation part 531 and a step for creating the SIP message body 511 at a SIP message body creation part 530 are performed, respectively.

As shown in Fig. 31, if it is desired to making a call with the UserID of the caller concealed from the callee, a flag indicating the SIP server that a Caller Anonymous Call is to be made is described in the SIP message header 560,

for example, a numeric 184 (561) attached to the position directly before the telephone number of callee in the case of making a Caller Anonymous Call at an exchange-based telephone system or extended header (571).

Then, the steps for creating a SIP Start Line INVITE indicating a request to the callee and creating the SIP message at a SIP signaling generation part 525 and the steps for creating the IP packet 60 and sending the DIP signal 512 at an IP packet processing part 526 are performed, respectively to initiate conversation.

At the end of the conversation, a step for erasing the account registration entry 514 is performed at the user data processing part 532 t erase the account from the SIP server and a step for discarding the IP address 515 is performed at the temporary IP address processing part 529, respectively, to complete the process.

The procedure for receiving the SIP message is the same as that for sending it with an exception that the step for obtaining the temporary address and the step for registering the account are performed when the IP Telephone is powered on or when the IP Telephone logs in the domain managed by the SIP server, the SIP message is received, conversation is made, the temporary IP address is discarded at the end of the conversation, and immediately after then,

a new temporary IP address is obtained for account registration.

As known from the description above, the IP Telephone has two temporary IP addresses, one for sending and one for receiving, while the IPv4 telephone set has either of them because two addresses cannot be set on one terminal at the same time.

The callee receiving the SIP signal from the SIP server with header conversion function according to the embodiment of the present invention described above can recognize that the caller makes a Caller Anonymous Call by checking the converted UserID indicating an anonymous in the SIP message.

If the DomainID in the SIP message has been converted or erased for concealing from other, the callee receiving the SIP message cannot known the caller's domain.

The malicious third party, even when receiving the SIP message sent by the caller, is difficult to guess the caller because the UserID is concealed.

The callers can be managed by any organization, for example, a communications carrier because the SIP server contains the conversion tables associating between real UserIDs and their other parameters.

With the IP Telephone using the temporary IP addresses according to the embodiment of the present invention, the

IP address is modified for each call, making difficult it for the malicious third party to guess the caller even when he/she intercepts the IP packet during conversation.

In addition, when the SIP message is sent through IPv6, the third party is difficult to guess the caller because many IP addresses are described in the same segment.